CLAIMS

What is claimed is:

1. A packet switched communications system having a dynamic voice jitter buffer for use with voice over Internet protocol (VoIP) packets comprising:

a source transmitting at least one VoIP packet;

at least one router for routing the VoIP packet to a specified destination;

a destination for receiving the at least one VoIP packet; and

wherein the VoIP packet operates to convey congestion information regarding the packet switched communications system to at least one buffer located at the destination.

2. A packet switched communications system as in claim 1, wherein the VoIP packet conveys congestion information comprising the steps of:

setting the time-to-live (TTL) field in the VoIP packet to a predetermined value;

decrementing the TTL value by one count as it traverses each respective router in the packet switched communications system;

calculating the number of routers the VoIP packet has passed through based on a final TTL value determined at the destination; and

adjusting the capacity of the at least one buffer at the destination based on the final TTL value in order to mitigate non-periodic receipt of incoming VoIP packets at the destination.

3. A packet switched communications system as in claim 2, further including the step of:

selecting a first, second or third capacity of the at least one buffer based upon the final TTL value.

4. A packet switched communications system as in claim 1, wherein the VoIP packet conveys congestion information comprising the steps of:

determining the speed upon which the VoIP packet has been received at the at least one router;

setting at least one field in the VoIP packet to indicate if the packet has traversed at least one previous router below a predetermined speed; and

adjusting the capacity of at least one buffer at the destination based upon recognition of the at least one field in order to mitigate non-periodic receipt of incoming VoIP packets at the destination.

5. A packet switched communications system as in claim 4, further including the step of:

setting the least one field within the VoIP packet with a congestion value based upon the speed of the originating link.

6. A packet switch communications systems claim 4, further including the step of:

setting the least one field within the VoIP packet with a congestion value based upon the speed of the destination link.

7. A packet switched communications system as in claim 1, further including the step of:

selecting a first, second or third capacity of the at least one congestion value.

8. A packet switched communications system as in claim 1, wherein the VoIP packet conveys congestion information comprising the steps of:

determining at the at least one router if a received packet has encountered at least one congested router;

setting at least one field in the VoIP packet indicating if the communications speed of a destination link is below a predetermined threshold; and

adjusting the capacity of at least one buffer at the destination based upon recognition of the at least one field in order to mitigate non-periodic receipt of incoming VoIP packets at the destination.

9. A packet switched communications system as in claim 8, further comprising the step of:

selecting a first, second or third capacity of the at least one buffer based upon a value set within the at least one field.

10. A method for adjusting the size of a jitter buffer for use in a voice over Internet protocol (VoIP) packet switched communications system comprising the steps of:

adjusting the time-to-live (TTL) field in a VoIP packet to a predetermined value at a source;

decrementing the TTL field by at least one count each time the VoIP packet traverse a router in the VoIP packet system;

reading the TTL field at a destination; and

adjusting the size of a jitter buffer based upon the TTL value in order to mitigate the effect of receipt of non-period VoIP packets at the destination.

11. A method for adjusting the size of a jitter buffer, as in claim 10:

wherein the jitter buffer is located at the destination.

12. A method for adjusting the size of a jitter buffer, as in claim 10 further includes the steps of:

comparing the predetermined value of the TTL field with the value read at the destination to produce a compared value; and

mapping the compared value to a predetermined jitter buffer capacity to provide a substantially continuous flow of VoIP packets from jitter buffer.

13. A method for adjusting the size of a jitter buffer as in claim 12, further comprising the step of:

setting the capacity of the jitter buffer to either a first, second or third predetermined capacity based upon the compared value.

14. A method for adjusting the size of a jitter buffer for use with a packet network transmitting voice over Internet protocol (VoIP) packets based upon transmission path delay comprising the steps of:

determining the amount of transmission delay through a transmission path that a VoIP packet has encountered upon receipt by at least one router in the packet network;

setting a field within the VoIP packet when the transmission rate for a link used for the VoiP is below a predetermined threshold;

recognizing the field at a destination of the VoIP packet; and

adjusting the size of a jitter buffer based upon recognition of the field in order to mitigate the effect of receipt of non-periodic VoiP packets at the destination.

15. A method for adjusting the size of a jitter buffer as in claim 14, wherein the jitter buffer is located at the destination.

16. A method for adjusting the size of a jitter buffer as in claim 14, further including the steps of:

setting the field using a numeric value based upon the amount of transmission path delay; and

mapping the numeric value into a minimal jitter buffer size required for that amount of delay.

17. A method for adjusting the size of a jitter buffer as in claim 14, further comprising the step of:

adjusting the size of the jitter buffer to either a first, second or third capacity based upon the numeric value set within the field.

18. A method for adjusting the size of a jitter buffer for use with a packet network transmitting voice over Internet protocol (VoIP) packets based upon transmission path delay comprising the steps of:

determining the amount of transmission delay through a transmission path that a VoIP packet has encountered upon receipt by at least one router in the packet network;

setting a field within the VoIP packet when the congestion of the link exceeds a predetermined threshold;

recognizing the field at a destination of the VoIP packet; and

adjusting the size of a jitter buffer based upon recognition of the field in order to mitigate the effect of receipt of non-periodic VoiP packets at the destination.

19. A method for adjusting the size of a jitter buffer as in claim 18, wherein the jitter buffer is located at the destination.

20. A method for adjusting the size of a jitter buffer as in claim 18, further including the steps of:

setting the field using a numeric value based upon the link congestion; and

mapping the numeric value into a minimal jitter buffer size required for that amount of delay.

21. A method for adjusting the size of a jitter buffer as in claim 18, further comprising the step of:

adjusting the size of the jitter buffer to either a first, second or third capacity based upon the numeric value set within the field.